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54. Title of the invention: Electroacoustic transducer

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Specification

1. **Title of the Invention**

Electroacoustic transducer

2. **Claims**

(1) An electroacoustic transducer characterised in that a signal from a source in the audible frequency band and a direct current are added in a fixed proportion and subjected to square root conversion and then multiplied with a carrier of a frequency adequately higher than the signal frequency and power-amplified, before being led to an ultrasonic vibrator, which performs electroacoustic conversion, and the signal thus modulated is converted to an acoustic wave with a finite amplitude level and radiated into air or other medium where the original audible sound is reproduced by the nonlinear effect of the medium.

(2) An electroacoustic transducer, according to Claim 1, characterised in that the said signal in the audible frequency band is passed though a double integrator beforehand so that it is integrated twice in time.

3. **Detailed Description of the Invention**

<u>Technical field</u>

The present invention relates to an electroacoustic transducer used to generate an electric signal in the audible frequency band in air as an audible signal.

<u>Prior art</u>

In the prior art, electrodynamic direct speakers and horn-loaded speakers have been the main forms of electroacoustic transducers and in both of these mechanical vibrational energy is converted to acoustic energy by vibrating a vibrator in air and creating a compression wave.

The means used in the present invention is completely different from the acoustic transducers of the speakers etc of the prior art in that it uses the parametric action of sound waves of a finite amplitude caused by the non-linear effect of air. The sound waves ('secondary waves') reproduced by self-demodulation in air due to the parametric

actions are characterised in having a directivity pattern similar to that of carrier sound waves in the ultrasonic region.

One previously-reported system used as a transmission system is the parametric speaker, in which ultrasonic waves amplitude-modulated by a signal in the audible frequency band are radiated into air, water or another medium at a finite amplitude level and the sound waves are reproduced in the medium by self-demodulation based on that medium's non-linearity effect. One of the characteristics of the parametric speaker, which exploits the non-linearity phenomenon of sound waves, is the sharpness of its directivity; in this, a high-frequency carrier wave is amplitude-modulated by a signal wave, which is an audible sound, and transmitted as a finite amplitude wave, and the secondary wave, which is related to the signal by the nonlinear interactions of the sound waves, is distributed in space as a vertical array, resulting in sharp directionality and a reduction in the size of the side lobe.

Let it be assumed that a finite amplitude wave with envelope f (t)

$$P_1 = P_0\, f(t) \sin \omega_0 t \quad \ldots\ldots\ldots\ldots(1)$$

is emitted by a circular transmitter of radius a. Here, P_0 is the sound-source sound pressure and ω_0 is the angular frequency of the carrier wave. If it is assumed that this primary wave is fully collimated by the planar wave, and the secondary wave P_2 is on the sonic axis then:

$$P_2 = \frac{\beta P_0^2 a^2}{16 \rho_0 C_0^4 \alpha r} \frac{\partial^2}{\partial t^2} f^2 \left(t - \frac{Z}{C_0}\right) \quad \ldots. (2)$$

Here, β is the non-linear parameter of the medium, ρ_0 is the medium density, C_0 is the acoustic velocity and α is the linear absorption coefficient of the primary wave. According to Formula (2), P_2 is proportional to f^2. Thus the secondary wave is produced by squaring of the envelope, a non-linear operation. This squaring operation, which is a direct result of secondary non-linearity of the sound waves, is a cause of distortion of the reproduced signal being generated. Now, if

$$f(t) = \sqrt{1 + S(t)} \qquad \ldots\ldots (3)$$

then P_l is proportional to the signal s (t) and no distortion is produced. This modulation will be referred to as modified double side band (MDSB).

Table 1 shows the dB, when a comparison is made between the properties of the single side band (SSB) design and modified double side band (MDSB) design, with the double side band (DSB) design as standard for a modulator in a parametric speaker.

Table 1

Design Item	DSB	SSB	MDSB
Second high-frequency distortion (%)	100	0	0
Reproduced sound pressure Power limited	0	-4.4	-2.5
Reproduced sound pressure amplitude limited	0	-1.0	0
Band (Hz) of electric signal	f_0-f to f_0 + f	f_0-f to f_0 or f_0 to f_0 + f	A wide band centred on f_0

Here, the case examined is one in which the signal is a sine wave, there is 100% modulation and, in Formula (3), $\sqrt{1 + \sin \omega_0 t}$. The results show that DSB is poor as to distortion but shows good conversion efficiency and MDSB is superior in not producing any distortion. However, because the electrical signal is broadband, the distortion is all the larger if a transmitter which does not cover this frequency range is used.

Figs 1 and 2 show, respectively, the experimental results for efficiency and distortion for the DSB and MDSB designs when ultrasonic vibrators with a central frequency of 40KHz are used in a planar arrangement of 581 pieces. Fig. 1 shows the second and third harmonics when the peak value of the voltage impressed on the speaker is set to a constant of 9.5m for the reproduced sound pressure distance and Fig. 2 shows these

when the fundamental wave is 0dB. Comparative tests for efficiency were conducted with the condition of 'amplitude limited', i.e. the peak value for the voltage impressed onto the speaker was fixed; in both designs a similar sound pressure was obtained, which accords to theory. As shown in Fig. 2, the distortion was approximately 10dB less with the MDSB design than with the DSB design, at least for secondary harmonics. However, since the vibrator was not broadband, there was a tendency for third harmonics to be larger.

The above designs have the fault that the second harmonic distortion component is great. In particular, the second harmonic distortion is directly related to the modulation factor in amplitude modulation and this deteriorates as the harmonic becomes deeper.

Purpose

The present invention has the object of preventing the faults described above i.e. the deterioration in secondary harmonic distortion characteristics, i.e. in a speaker (a 'parametric' speaker) in which ultrasonic waves amplitude modified by a signal in the audible frequency band is emitted into the air with a finite amplitude level, and a self-modulated audible signal is obtained by the nonlinear effects of air.

Constitution

The constitution of the present invention is described below with reference to embodiments.

Generally, as the frequency of an ultrasonic wave becomes greater, a sound wave emitted by a vibrator is propagated in the form of a beam.

Assuming that an amplitude-modulated ultrasonic wave from a vibrator array of radius a is emitted in a beam, the sound pressure P at a point at a distance x from the array is expressed by the following formula.

$$P = P_0 \{1 + m \cdot g(t - \frac{x}{C_0})\} e^{-\alpha x} \sin(\omega_0 t - k_0 x) \quad \text{……… (4)}$$

Here, C_0 is the sound velocity, α is the damping coefficient of the sound velocities of frequencies ω_0, P_0 is the initial sound pressure, m is the modulation factor, and g(t) is the modulating wave. The sound pressure of the secondary wave produced when the

finite amplitude level ultrasonic wave shown in Formula (3) is demodulated by nonlinear parametric action in air is expressed by the following non-homogeneous wave equation.

$$\nabla^2 P_s - \frac{1}{C_0^2} \cdot \frac{\partial^2 P_s}{\partial t^2} = -\rho_0 \frac{\partial q}{\partial t}$$

……… (5)

In Equation 5, P_S is the sound pressure of the secondary wave, ρ_0 is the air density, q is the hypothetical sound-source density of the secondary wave produced in the primary wave beam and q is expressed by the following equation

$$q = \frac{P}{\rho_0^2 C_0^4} \cdot \frac{\partial}{\partial t} \rho^2 \qquad \text{……… (6)}$$

Therefore, if the hypothetical sound-source density is calculated at a point at distance x (axial) from the array, using equations (4) and (6), the following equation is obtained.

$$q = \frac{\beta P_0^2}{\rho_0^2 C_0^4} e^{-2\alpha x} \frac{\partial}{\partial t} \left[m \, . \, g \left(t - \frac{x}{C_0}\right) + \frac{1}{2} m^2 g^2 \left(t - \frac{x}{C_0}\right) \right]$$

……… (7)

In Equation (7), based on the components of the signal, the first term on the right expresses the hypothetical sound-source density and the second term expresses the hypothetical sound-source density of the distortion component.

As described above, the present invention relates to a modulation method to remove the distortion component produced in audio transducers that use nonlinear parametric action. Thus, it is a modulation method in which the direct current component in the

modulating signal is added and square root converted and then the sum of this and the carrier is found.

In this case, the modulating signal is expressed by the following equation.

$$V = \sqrt{1 + m \cdot g(t)}\ \sin \omega_0 t \qquad \ldots\ldots\ldots (8)$$

Therefore, the sound pressure of the primary wave (the modulated ultrasonic wave) at a point at distance x from the vibrator array is:

$$P = P_0 \sqrt{1 + m \cdot g\left(t - \frac{x}{C_0}\right)}\ e^{-\alpha x} \sin(\omega_0 t - k_0 x) \qquad \ldots\ldots\ldots (9)$$

In this case, the virtual sound-source density of the secondary wave, found using Equation (6), is:

$$q = \frac{\beta P_0^2}{2\rho_0^2 C_0^4} e^{-2dx} \cdot m \cdot \frac{\partial}{\partial t} g\left(t - \frac{x}{C_0}\right) \qquad \ldots\ldots\ldots (10)$$

Therefore, if this modulation method shown in term 2 on the right of Equation (7) is used, it can be expected that the distortion component will be eradicated and the quality of the sound reproduction markedly improved.

[Embodiment 1)

Fig. 3 shows an example of the basic constitution required to carry out the method according to the invention. In Fig. 3, 1 is a modulation signal source (audible frequency band), 2 is a coefficient multiplier, 3 is a direct current source, 4 is an adder, 5 is a square root converter, 6 is an ultrasonic wave band oscillator, 7 is a multiplier, 8 is a power amplifier and 9 is an ultrasonic vibrator array.

(Embodiment 2)

Fig. 4 shows an example of a modified constitution required to carry out the method according to the invention. In Fig. 4, 10 is a double integrator and all other parts are identical with Fig. 3. The reproduced sound pressure obtained using a nonlinear parametric speaker based on this modulation method, at a point x on the axis of the array, is given finally as:

$$P_s = \frac{\beta P_0^2 a^2 m}{16\rho_0 C_0^4 \alpha x} \frac{\partial^2}{\partial t^2} g\left(t - \frac{x}{C_0}\right) \qquad \ldots\ldots\ldots(11)$$

Thus the reproduced sound pressure is proportional to the second-degree differential of the original modulating signal. Therefore it is possible to obtain a reproduced sound pressure proportional to the original modulating signal by passing the modulating signal through a double integrator prior to modulation, as shown in Fig. 4. This may be expressed as

$$P_s = \frac{\beta P_0^2 a^2 m}{16\rho_0 C_0^4 \alpha x} g\left(t - \frac{x}{C_0}\right) \qquad \ldots\ldots\ldots(12)$$

Effects

As is clear from the description above, when the method according to the invention is used, the harmonic distortion of the reproduced sound is improved and high-quality sound reproduction is achieved. In methods of the prior art, when the modulation degree ($m \leq 1$) is made deeper, the distortion became markedly worse but, with the method according to the invention, since, basically, distortion can be reduced with no relationship to m, this effect is most marked when m is large (i.e. $m \leq 1$). Further, since the reproduced sound pressure is proportional to m, the fact that it is possible to use a

large m is also of great benefit in terms of the improvement of the performance of the audio transducer.

4. Simple Description of the Figures

Fig. 1 is an efficiency comparison figure for ultrasonic vibrators; Fig. 2 is a distortion comparison figure; Figs 3 and 4 are, respectively, electrical circuit diagrams to describe embodiments of the invention

1 … modulating signal source; 2 … coefficient multiplier; 3 … DC source; 4 … adder; 5 … square root converter; 6 … ultrasonic waveband oscillator; 7 … multiplier; 8 … power amplifier; 9 … ultrasonic vibrator array; 10 … double integrator.

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